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Amendment Dated 28 December 2004
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Amendments to the Claims

This listing of claims will replace all prior versions, and listings, of claims in the application:

Listing of Claims:

Claims 1-49 (Withdrawn)

Claims 50-59 (Canceled).

Claim 60 (Amended):

A system for measuring quality of voice services on modern telephony networks, including Voice Over Network (VON), Public Switched Telephone Network (PSTN), and hybrid VON/PSTN networks, comprising:

an analyzer that measures quality of service of a signal transmitted over the telephony network;

wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; and

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol;

~~The system of claim 59,~~

and wherein said analyzer further comprises a report generator that reports ~~both~~ a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

Claim 61 (Amended): The system of claim 60 ~~51,~~ wherein said analyzer further comprises~~:~~ a computer system; and a configuration subsystem that configures said computer system to interface with said telephony network~~; a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; and a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol~~.

Claim 62 (Original): The system of claim 61, wherein said system is a personal computer system comprising hardware and software components.

Claim 63 (Original): The system of claim 61, wherein said connector is a handset connector that couples to a telephone handset to make said connection.

Claim 64 (Original): The system of claim 61, wherein said connector is a base connector that couples to a telephone base to make said connection.

Claim 65 (Original): The system of claim 61, further comprising a recorder that records signals transmitted by the telephony network at said test point.

Claim 66 (Original): The system of claim 65, further comprising a reproduction unit that reproduces the recorded signal through the telephony network at said test point to permit a determination of a speech quality.

Claim 67 (Original): The system of claim 61, wherein said quality of service analysis subsystem comprises a subsystem that tests quality of service selected from a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

Claim 68 (Original): The system of claim 61, wherein said quality of service analysis subsystem comprises a subsystem that tests a telephony condition selected from an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

Claim 69 (Original): The system of claim 61, wherein said connector comprises a connector adapted to connect to a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 70 (Original): The system of claim 61, further comprising a comparison subsystem that compares results obtained from tests performed between two or more locations in said telephony network.

Claim 71 (Original): The system of claim 70, wherein said signals comprise signals selected from one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

Claim 72 (Original): The system of claim 70, wherein said signals comprise a signal obtained from a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 73 (Original): The system of claim 70, wherein said signals comprise a signal obtained from a telephone handset and a signal obtained from a non-telephone device connected to a communication channel.

Claim 74 (Original): The system of claim 61, further comprising a recording subsystem that records a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

Claim 75 (Original): The system of claim 74, wherein said recording subsystem comprises a nonvolatile memory that maintains a record of a selected one of a DC

impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

Claim 76 (Original): The system of claim 61, further comprising a reproduction subsystem that reproduces from a record a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

Claim 77 (Original): The system of claim 61, further comprising a synchronizing subsystem that synchronizes a first selected one of a waveform, a timing signal, or a line event with a second selected one of a waveform, a timing signal, or a line event.

Claim 78 (Original): The system of claim 61, wherein the computer system comprises a modeling subsystem that models mathematically a parameter of a programmable telephone equipment, said parameter selected from one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

Claim 79 (Original): The system of claim 61, further comprising: a transmitter subsystem that sends over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, all AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

Claim 80 (Original): The system of claim 79, further comprising a tone and waveform generator that sends tones and selected waveforms over the telephony network.

Claim 81 (Original): The system of claim 79, further comprising a synchronizing subsystem that synchronizes a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

Claim 82 (Original): The system of claim 79, further comprising a recorder that records speech and signals.

Claim 83 (Original): The system of claim 79, further comprising a digitizer that digitizes recorded speech and signals.

Claim 84 (Original): The system of claim 79, further comprising a scheduler that commands the transmitter to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording the time of transmission of said file so that a delay and a degradation of said signal in one-way transmission can be analyzed.

Claim 85 (Original): The system of claim 79, further comprising: a recorder that records speech and signals; a digitizer that digitizes recorded speech and signals; and a scheduler that commands the transmitter to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording the time of transmission of said file, so that a delay and a degradation of said signal in one-way transmission can be analyzed.

Claim 86 (Original): The system of claim 79, further comprising: a second transmitter that can generate a second call using a second network; a second analyzer that measures quality of service of a signal transmitted over said second network, said second analyzer comprising: a computer system; a configuration subsystem that configures said computer system to interface with said second network; an quality of service analysis subsystem that performs an analysis of said quality of service of said signal; a report generator that reports a result of said analysis; and a comparator that compares the relative performance of the telephony network and said second network.

Claim 87 (Original): The system of claim 86, wherein said second transmitter comprises a transmitter that repeatedly generates said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

Claim 88 (Original): The system of claim 61, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

Claim 89 (Original): The system of claim 61, further comprising: a receiver subsystem that receives over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone and a CLASS signal.

Claim 90 (Original): The system of claim 89, further comprising a receiver that detects and recognizes tones and selected waveforms transmitted over the telephony network.

Claim 91 (Original): The system of claim 89, further comprising a synchronization detector that determines a synchronization between a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

Claim 92 (Original): The system of claim 89, further comprising a speech measurement subsystem that measures quality of service of a signal representative of speech.

Claim 93 (Original): The system of claim 89, further comprising a signal measurement subsystem that measures signal levels.

Claim 94 (Original): The system of claim 89, further comprising an analyzer that detects and analyzes DTMF signals.

Claim 95 (Original): The system of claim 89, further comprising a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

Claim 96 (Original): The system of claim 89, further comprising: a speech measurement subsystem that measures quality of service of a signal representative of speech; a signal measurement subsystem that measures signal levels; an analyzer that detects and analyzes DTMF signals; and a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

Claim 97 (Original): The system of claim 89, further comprising: a second receiver that can accept a second call transmitted over a second network; a second analyzer that measures quality of service of a signal transmitted over said second network

Claim 98 (Original): The system of claim 97, wherein said second analyzer comprises a comparator that compares the relative performance of the telephony network and said second network.

Claim 99 (Original): The system of claim 97 wherein said second receiver comprises an receiver that repeatedly receives said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

Claim 100 (Original): The system of claim 89, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

Claims 101-108 (Canceled).

Claim 109 (Amended):

A system for analyzing adherence to a communication protocol, comprising:

a connector adapted to make a connection to a test point in the telephony network;

an analyzer that analyzes adherence to a communication protocol at said test point, said analyzer further comprising:

a computer system;

a configuration subsystem that configures said computer system to interface with said telephony network;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol;

and a report generator that reports a result of said communication protocol analysis; and

~~The system of claim 101, further comprising~~ a comparison subsystem that compares adherence to a communication protocol as used in two or more locations in said telephony network.

Claim 110 (Original): The system of claim 109, wherein said communication protocol is selected from one of an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

Claim 111 (Original): The system of claim 109, where in said communication protocol is analyzed from a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 112 (Original): The system of claim 111, wherein said communication protocol is analyzed from a telephone handset and from a non-telephone device connected to a communication channel.

Claim 113 (Original): The system of claim 109 ~~101~~, further comprising a recording subsystem that records the analysis of the communication protocol.

Claim 114 (Original): The system of claim 113, further comprising a non-volatile memory that maintains a record of the analysis of the communication protocol.

Claim 115 (Original): The system of claim 114, further comprising a reproduction subsystem that reproduces from a record the analysis of the communication protocol.

Claim 116 (Original): The system of claim 113, wherein the computer system comprises a modeling subsystem that models mathematically the communication protocol used by a programmable telephone equipment, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

Claims 117-126 (Canceled).

Claim 127 (Amended):

A system for measuring quality of voice services on modern telephony networks, including Voice Over Network (VON), Public Switched Telephone Network (PSTN), and hybrid VON/PSTN networks, and for analyzing adherence to a communication protocol, comprising:

an analyzer that measures quality of service of a signal transmitted over the telephony network and analyzes adherence to a communication protocol at a test point in the telephony network;

wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; and

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol;

~~The system of claim 126,~~ wherein said analyzer further comprises a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

Claim 128 (Amended):

A system for measuring quality of voice services on modern telephony networks, including Voice Over Network (VON), Public Switched Telephone Network (PSTN), and hybrid VON/PSTN networks, and for analyzing adherence to a communication protocol, comprising:

an analyzer that measures quality of service of a signal transmitted over the telephony network and analyzes adherence to a communication protocol at a test point in the telephony network;

~~The system of claim 117,~~ wherein said analyzer comprises: a computer system; a configuration subsystem that configures said computer system to interface with said telephony network; a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol; and a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

Claim 129 (Original): The system of claim 128, wherein said system is a personal computer system comprising hardware and software components.

Claim 130 (Original): The system of claim 128, wherein said connector is a handset connector that couples to a telephone handset to make said connection.

Claim 131 (Original): The system of claim 128, wherein said connector is a base connector that couples to a telephone base to make said connection.

Claim 132 (Original): The system of claim 128, further comprising: a recorder that records signals transmitted by the telephony network at said test point.

Claim 133 (Original): The system of claim 132, further comprising a reproduction unit that reproduces the recorded signal through the telephony network at said test point to permit a determination of a speech quality.

Claim 134 (Original): The system of claim 128, wherein said quality of service analysis subsystem comprises a subsystem that tests quality of service selected from a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

Claim 135 (Original): The system of claim 128, wherein said quality of service analysis subsystem comprises a subsystem that tests a telephony condition selected from an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

Claim 136 (Original): The system of claim 128, wherein said communication protocol analysis subsystem comprises a subsystem that tests an adherence to a communication protocol selected from an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

Claim 137 (Original): The system of claim 128, wherein said connector comprises a connector adapted to connect to a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 138 (Original): The system of claim 128, further comprising a comparison subsystem that compares results obtained from tests performed between two or more locations in said telephony network.

Claim 139 (Original): The system of claim 138, wherein said signals comprise signals selected from one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

Claim 140 (Original): The system of claim 138, wherein said signals comprise a signal obtained from at least a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 141 (Original): The system of claim 138, wherein said signals comprise a signal obtained from a telephone handset and at least a signal obtained from a non-telephone device connected to a communication channel.

Claim 142 (Original): The system of claim 128, further comprising a recording subsystem that records a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

Claim 143 (Original): The system of claim 128, further comprising a non-volatile memory that maintains a record of a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

Claim 144 (Original): The system of claim 128, further comprising a reproduction subsystem that reproduces from a record a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

Claim 145 (Original): The system of claim 128, further comprising a synchronizing subsystem that synchronizes a first selected one of a waveform, a timing signal, or a line event with a second selected one of a waveform, a timing signal, or a line event.

Claim 146 (Original): The system of claim 128, wherein the computer system comprises a modeling, subsystem that models mathematically a parameter of a programmable telephone equipment, said parameter selected from one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

Claim 147 (Original): The system of claim 128, further comprising a tone and waveform generator that sends tones and selected waveforms over the telephony network.

Claim 148 (Original): The system of claim 128, further comprising a synchronizing subsystem that synchronizes a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

Claim 149 (Original): The system of claim 128, further comprising a recorder that records speech and signals.

Claim 150 (Original): The system of claim 128, further comprising a digitizer that digitizes recorded speech and signals.

Claim 151 (Original): The system of claim 128, further comprising a transmitter subsystem that sends over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

Claim 152 (Original): The system of claim 151, further comprising a scheduler that commands the transmitter to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording the time of transmission of said file, so that a delay and a degradation of said signal in one-way transmission can be analyzed.

Claim 153 (Original): The system of claim 151, further comprising: a second transmitter that can generate a second call using a second network; a second analyzer that measures quality of service of a signal transmitted over said second network and analyzes adherence to a communication protocol at said test point

Claim 154 (Original): The system of claim 153, wherein said second analyzer comprises a comparator that compares the relative performance of the telephony network and said second network.

Claim 155 (Original): The system of claim 154, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

Claim 156 (Original): The system of claim 155 further comprising a billing subsystem that records billing information pertaining to said route selected by said routing selector.

Claim 157 (Original): The system of claim 153, wherein said second transmitter comprises a transmitter that repeatedly generates said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

Claim 158 (Original): The system of claim 157, further comprising a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place.

Claim 159 (Original): The system of claim 158, further comprising a billing subsystem that records billing information pertaining to said new route selected.

Claim 160 (Original): The system of claim 128, further comprising a receiver subsystem that receives over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone and a CLASS signal.

Claim 161 (Original): The system of claim 160, further comprising a receiver that detects and recognizes tones and selected waveforms transmitted over the telephony network.

Claim 162 (Original): The system of claim 161, further comprising a synchronization detector that determines a synchronization between a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

Claim 163 (Original): The system of claim 160, further comprising a speech measurement subsystem that measures quality of service of a signal representative of speech.

Claim 164 (Original): The system of claim 160, further comprising a signal measurement subsystem that measures signal levels.

Claim 165 (Original): The system of claim 160, further comprising an analyzer that detects and analyzes DTMF signals.

Claim 166 (Original): The system of claim 160, further comprising a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

Claim 167 (Original): The system of claim 160, further comprising: a speech measurement subsystem that measures quality of service of a signal representative of speech; a signal measurement subsystem that measures signal levels; an analyzer that detects and analyzes DTMF signals; and a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

Claim 168 (Original): The system of claim 160, further comprising a second receiver that can accept a second call transmitted over a second network.

Claim 169 (Original): The system of claim 168, further comprising a second analyzer that measures quality of service of a signal transmitted over said second network and analyzes adherence to a communication protocol at said test point.

Claim 170 (Original): The system of claim 169, further comprising a comparator that compares the relative performance of the telephony network and said second network.

Claim 171 (Original): The system of claim 170 wherein said second receiver comprises an receiver that repeatedly receives said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

Claim 172 (Original): The system of claim 171, further comprising a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place.

Claim 173 (Original): The system of claim 160, further comprising a routing, selector that selects a route based at least in part on said quality of service measurement.

Claim 174 (Original): The system of claim 128, further comprising a billing subsystem that records billing information pertaining to the communications network.

Claim 175 (Original): The system of claim 128, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

Claim 176 (Original): The system of claim 175 further comprising a billing subsystem that records billing information pertaining to said route selected by said routing selector.

Claim 177 (Original): The system of claim 128, further comprising a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place.

Claim 178 (Original): The system of claim 177, further comprising a billing subsystem that records billing information pertaining to said new route selected.

Claim 179-185 (Withdrawn)

Claim 186 (Original): A system for measuring quality of voice services on modem telephony networks, including VON, PSTN, and hybrid VON/PSTN networks, and for analyzing adherence to a communication protocol, said system comprising:

a connector adapted to make a connection to a test point in the telephony network, said connector adapted to connect to at least a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router;

an analyzer that measures quality of service of a signal transmitted over the telephony network and analyzes adherence to a communication protocol at said test point, said analyzer comprising:

a computer system;

a configuration subsystem that configures said computer system to interface with said telephony network;

a tone and waveform generator that sends tones and selected waveforms over the telephony network; and

a synchronizing subsystem that synchronizes a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event;

a transmitter subsystem that sends over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal;

a scheduler that commands the transmitter subsystem to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording a time of transmission of said file so that a delay and a degradation of said signal in one-way transmission can be analyzed;

a receiver that detects and recognizes tones and selected waveforms transmitted over the telephony network and comprising a receiver subsystem that receives over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal;

a synchronization detector that determines a synchronization between a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event;

a recorder that records speech and signals transmitted by the telephony network at said test point, said recorder comprising non-volatile memory;

a recording subsystem that records a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said recording subsystem comprising a non-volatile memory that maintains a record of at least one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event;

a reproduction unit that reproduces recorded signal through the telephony network at said test point to permit a determination of a speech quality, said reproduction unit further comprising a reproduction subsystem that reproduces from a record a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event;

a modeling subsystem that models mathematically a parameter of a programmable telephone equipment, said parameter selected from one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment;

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal, said quality of service analysis subsystem comprising: (a) a subsystem that tests quality of service selected from a figure of merit for voice clarity, voice intelligibility, a voice level, a power, a timing signal, or a delay; and (b) a subsystem that tests a telephony condition selected from an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol, said communications protocol analysis subsystem comprising a subsystem that tests an adherence to a communication protocol selected from an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, or voice over cable;

a speech measurement subsystem that measures quality of service of a signal representative of speech;

a signal measurement subsystem that measures signal levels;

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol;

a signal analyzer that detects and analyzes DTMF signals;

a display subsystem that reports a selected result of at least one from quality of service measurement, signal level measurement, or signal type analysis; and

a comparison subsystem that compares results obtained from tests performed between two or more locations in said telephony network, wherein: (a) said signals comprise signals selected from one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay; (b) one of said signals comprise a signal obtained from a telephone handset; and (c) a second of said signals comprise a signal obtained from a selected one of a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, a router, or from a non-telephone device connected to a communication channel;

a second analyzer that measures quality of service of a signal transmitted over said second network and analyzes adherence to a communication protocol at said test point wherein said second analyzer repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place, said second analyzer comprising:

> a second transmitter that can generate a second call using a second network wherein said second transmitter comprises a transmitter that repeatedly generates said test calls; and
>
> a second receiver that can accept a second call transmitted over a second network wherein said second receiver comprises a receiver that repeatedly receives said test calls;

a routing selector that selects a route based at least in part on said quality of service measurement;

a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place;

a billing subsystem that records billing information pertaining to at least one selected from a route selected by said routing selector, a reroute selected, or other carrier billing information;

a connector adapted for a non-intrusive, high-impedance interface to said telephony network;

a high-impedance measurement subsystem capable of measuring quality of voice services passing through each said node on each said telephony network;

a quality of service analysis subsystem that analyzes quality of service for each said telephony network for a plurality of calls made over said telephony network through each said node;

a data store that records the analysis of the quality of service analysis subsystem;

a reporting subsystem that sends one or more said analyses of the quality of service analysis subsystem to a predetermined destination via the telephony network;

a monitoring subsystem that analyzes said recorded analyses and ascertains whether said quality of service falls below a predefined level; and

a notification subsystem whereby said analyzer system sends a message via the telephony network to a predefined destination when said quality of service falls below a predefined level;

at least one high-impedance measurement subsystem capable of measuring quality of service of each said telephony network by analyzing a voice communication passing through each said node on each said telephony network;

at least one quality of service analysis subsystem that analyzes quality of service for at least one of said telephony networks for the duration of at least one call made over said telephony networks and passing through said corresponding nodes connected to said telephony networks;

at least one switching subsystem corresponding to each quality of service analysis subsystem that, if during the course of a call said quality of service analysis subsystem detects a degradation of said quality of service on a first telephony network, said switching subsystem immediately switches the call to a second telephony network for uninterrupted servicing of said call;

at least two analyzer systems wherein each such analyzer system is connected to a node on at least two of said telephony networks, each such analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

at least one testing subsystem corresponding to one said analyzer system by which such analyzer system initiates at least two calls over at least two of said telephony networks to at least one other analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said telephony networks over which said calls were made;

at least one selection subsystem corresponding to each testing subsystem by which said corresponding analyzer system can select one telephony network from among the plurality of telephony networks tested for routing at least one call through said node corresponding to said corresponding analyzer system to another node, said selection based at least in part on an evaluation of said analyses by said corresponding testing subsystem; and

at least one circuit-switched network wherein said analyzer system is connected to at least one node corresponding to said circuit-switched network and wherein said system can select said circuit-switched network in lieu of said telephony networks.

Claim 187 (New):

A method that measures quality of voice services on modern telephony networks, including Voice Over Network (VON), Public Switched Telephone Network (PSTN), and hybrid VON/PSTN networks and for analyzing adherence to a communication protocol, comprising:

measuring the quality of service of a signal transmitted over the telephony network and analyzing the adherence to a communication protocol at a test point in the telephony network with an analyzer;

wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol; and

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

Claim 188 (New): The method of claim 187 wherein said communication protocol is selected from one of an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

Claim 189 (New): The method of claim 187 where in said communication protocol is analyzed from a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 190 (New): The method of claim 187 further comprising a recording subsystem that records the analysis of said communication protocol.

Claim 191 (New): The method of claim 187 further comprising a modeling subsystem that models mathematically the communication protocol used by a programmable telephone equipment, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

Claim 192 (New):

A method to manufacture an analyzer that measures quality of voice services on modern telephony networks, including Voice Over Network (VON), Public Switched Telephone Network (PSTN), and hybrid VON/PSTN networks and for analyzing adherence to a communication protocol, comprising:

providing an analyzer that measures the quality of service of a signal transmitted over the telephony network and analyzes the adherence to a communication protocol at a test point in the telephony network;

wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol; and

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

Claim 193 (New): The method of claim 192 wherein said communication protocol is selected from one of an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

Claim 194 (New): The method of claim 192 where in said communication protocol is analyzed from a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 195 (New): The method of claim 192 further comprising a recording subsystem that records the analysis of said communication protocol.

Claim 196 (New): The method of claim 192 further comprising a modeling subsystem that models mathematically the communication protocol used by a programmable telephone equipment, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

Claim 197 (New):

A program storage device readable by a computer that tangibly embodies a program of instructions executable by the computer to perform a method that measures quality of voice services on modern telephony networks, including Voice Over Network (VON), Public Switched Telephone Network (PSTN), and hybrid VON/PSTN networks and for analyzing adherence to a communication protocol, comprising:

measuring the quality of service of a signal transmitted over the telephony network and analyzing the adherence to a communication protocol at a test point in the telephony network with an analyzer;

wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol; and

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

Claim 198 (New): The program storage device of claim 197 wherein said communication protocol is selected from one of an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

Claim 199 (New): The program storage device of claim 197 where in said communication protocol is analyzed from a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

Claim 200 (New): The program storage device of claim 197 further comprising a recording subsystem that records the analysis of said communication protocol.

Claim 201 (New): The program storage device of claim 197 further comprising a modeling subsystem that models mathematically the communication protocol used by a programmable telephone equipment, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.